IN THE CLAIMS:

1. **(currently amended)** A voice coding method based on analysis-by-synthesis vector quantization comprising:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably replacing a position of a sample of the non-zero amplitude value in the configuration variable code book using only an index and a transmission parameter indicating a feature amount of voice without any additional supplementary information;

wherein the position and amplitude of the non-zero amplitude values coding an input speech signal are selected as an optimum series from entries in the configuration variable code book, which entries are varied by a certain rule rather than being determined from the input speech signal, and

wherein the number of non-zero amplitude values coding an input speech signal remains constant even if one or more of a lag value and a frame length of the input speech signal change.

2. **(previously presented)** The method according to claim 1, further comprising :

variably replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

3. **(previously presented)** The method according to claim 2, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable codebook within a region corresponding to the lag value

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depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

4. (previously presented) The method according to claim 1, further comprising:

variably replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

5. (previously presented) The method according to claim 4, further comprising:

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reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

6. (previously presented) The method according to claim 5, further comprising:

reconstructing the position of the sample the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on the pitch gain value.

7. (currently amended) A voice decoding method for decoding a voice signal coded by a voice coding method based on analysis-by-synthesis vector quantization comprising:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably replacing a position of a sample of the non- zero amplitude value in the configuration variable code book using only an index and a transmission parameter indicating a feature amount of voice without any additional supplementary information;

wherein the position and amplitude of the non-zero amplitude values coding the voice signal are selected as an optimum series from entries in the configuration variable codebook, which entries are varied by a certain rule rather than being determined from the voice signal, and

wherein the number of non-zero amplitude values coding an input speech signal remains constant even if one or more of a lag value and a frame length of the input speech signal change.

8. (previously presented) The method according to claim 7, further comprising: variably replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

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9. (previously presented) The method according to claim 8, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a ceding unit of the voice.

10. (previously presented) The method according to claim 7, further comprising:

variably replacing the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a

pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

11. (previously presented) The method according to claim 10, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

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12. (previously presented) The method according to claim 11, further comprising:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on the pitch gain value.

13. (currently amended) A voice coding apparatus based on analysis-by-synthesis vector quantization comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality non-zero amplitude values, wherein

said configuration variable code book unit variably replaces a position of a sample of the non-zero amplitude value in said configuration variable code book unit using only an index and a transmission parameter indicating a feature amount of voice without any additional supplementary information;

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wherein the position and amplitude of the non-zero amplitude values coding an input speech signal are selected as an optimum series from entries in the configuration variable codebook, which entries are varied by a certain rule rather than being determined from the input speech signal, and

wherein the number of non-zero amplitude values coding an input speech signal remains constant even if one or more of a lag value and a frame length of the input speech signal change.

14. (previously presented) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably replaces the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

15. (previously presented) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably replaces the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

16. (currently amended) A voice decoding apparatus for decoding a voice signal coded by a voice coding apparatus based on analysis-by-synthesis vector quantization comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality of non-zero amplitude values, wherein

said configuration variable code book unit variably replaces a position of a sample of the non-zero amplitude value using only an index and a transmission parameter indicating a feature amount of voice without any additional supplementary information;

wherein the position and amplitude of the non-zero amplitude values coding the voice signal are selected as an optimum series from entries in the configuration variable codebook, which entries are varied by a certain rule rather than being determined from the voice signal, and

wherein the number of non-zero amplitude values coding an input speech signal remains constant even if one or more of a lag value and a frame length of the input speech signal change.

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17. (previously presented) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably replaces the position of the sample of the non-hero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

18. (previously presented) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably replaces the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.